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METHOD OF FINE SYNCHRONIZATION TO A SIGNAL RECEIVED FROM A TRANSMISSION CHANNEL

The present invention relates to a method of fine synchronization to a receive signal corresponding to a reference signal transmitted in a transmission channel.

In a transmission system, in particular a radio transmission system, a transmitter transmits a reference signal to a receiver in a transmission channel. One of the first operations that the receiver must perform is to synchronize to the receive signal. This problem is well-known to the skilled person and it is therefore unnecessary to remind the reader here of the various techniques that have been used to obtain such synchronization.

If the signals are digital signals, it is usual to evaluate the synchronization error by means of a time unit which is the time difference between two successive bits of a signal and which is referred to as the bit period. It appears that the available prior art solutions cannot achieve synchronization to an accuracy much better than one bit period.

This accuracy can prove insufficient in some cases. This is because synchronization gives the transit time of the signal in the transmission channel, in other words the transmission time between the transmitter and the receiver. This data is important in a duplex radiocommunications system in which a base station communicates with a terminal because the base station and the terminal have respective transceivers and the terminal must operate in such a way that the signal it transmits arrives at a precise time with reference to the clock of the base station. To achieve this, it is naturally necessary for the terminal to know the time taken for the signal to reach the base station.

The transmission time is also a direct reflection of the distance between the transmitter and the receiver. Clearly, the accuracy of this distance is of fundamental

importance when it is a question of locating the terminal by identifying its position relative to one or more base stations. The general problem of locating a terminal is a very current concern because of its applications, which include cell handover strategies in cellular networks, for example. The security field should also be mentioned, whether in connection with identifying the geographical location of the source of an emergency call or the position of a stolen vehicle equipped with the terminal.

An object of the present invention is a synchronization method whose accuracy is much better than one bit period.

In accordance with the invention, the method of fine synchronization to a receive signal corresponding to a reference signal transmitted in a transmission channel includes the following steps:

- selecting a source signal producing a characterization signal after it has passed through said transmission channel,
- establishing a characterization matrix for estimating the covariance of the characterization signal,
- identifying dominant eigenvalues which are the highest eigenvalues of the characterization matrix,
- calculating the correlation function of the source signal with the sum of the eigenvectors associated with the dominant eigenvalues, and
- searching for the first maximum of the correlation function.

If the time increment adopted for calculating the correlation function is made sufficiently small, and in all cases much less than one bit period, the above method achieves very good accuracy.

A first option is for the number of dominant eigenvalues to be predetermined. This number typically represents 20% to 30% of the dimension of the characterization matrix.

A second option is for the ratio of the sum of the dominant eigenvalues to the sum of the all the eigenvalues to be greater than or equal to a predetermined percentage. In this case the percentage adopted is often greater than 90%, for example 95%.

A third option is for the method also to include a step of estimating the additive noise in the transmission channel, the dominant eigenvalues being such that their sum is less than or equal to the sum of all the eigenvalues less the additive noise.

Also, the additive noise is estimated by normalizing the instantaneous noise, which is evaluated from the receive signal, the reference signal and an estimate of the impulse response of the transmission channel.

The expression for the instantaneous noise is advantageously: $N_0 = S - A.X$, where A is the transmission matrix associated with the reference signal.

Whichever of the above options is adopted, the characterization matrix results from a smoothing operation.

In a preferred embodiment, the characterization signal is an estimate of the impulse response of the transmission channel.

The characterization signal can instead be the receive signal.

The present invention emerges in more detail from the following description of proposed embodiments of the invention, which is given by way of illustrative example and with reference to the accompanying figures, in which:

- Figure 1 shows a first variant of the invention, and
- Figure 2 shows a second variant.

Elements common to both figures are allocated the same reference numbers.

The receiver has already achieved coarse synchronization to the receive signal, to an accuracy of the order of one bit period, using any of the available prior art solutions.

The receive signal corresponds to a reference signal produced by the transmitter and known to the receiver. The reference signal can be known *a priori*, i.e. it can be a training sequence made up of identified symbols. It can also be determined *a posteriori* using techniques referred to generically as blind probing. In this case, during the synchronization procedure, the receiver regenerates from the receive signal the series of symbols forming the reference signal.

It is first necessary to characterize transmission between the transmitter and the receiver. To this end, a source signal produced by the transmitter is selected which yields a characterization signal in the receiver after transmission over the channel.

Of course, if the source signal is the reference signal, the characterization signal is the receive signal itself. This is not always the optimum solution, however, in terms of the complexity and performance of the method of the invention.

Another solution is to retain a modulated pulse as the source signal, in which case the characterization signal is the impulse response of the transmission channel.

For example, the GSM digital cellular mobile telephone system uses a training sequence made up of 26 symbols, the impulse response being generally estimated with five coefficients since it is accepted that the dispersion of the channel is equal to 4.

In this case, the receive signal has a maximum dimension of 22, which is significantly larger than that of the impulse response.

Two embodiments of the invention are therefore examined in succession and with reference to Figure 1, starting with the situation in which the source signal is a modulated pulse.

Estimating the impulse response does not give rise to any problems in itself because many methods are

available for this, for example the least squares method, which is described in particular in patent applications FR 2 696 604 and EP 0 564 849. Briefly, that technique uses a measurement matrix A constructed from the training sequence TS of length _n_. The matrix A has (n-d) rows and (d+1) columns, where _d_ represents the dispersion of the channel. The element in the ith row and the jth column is the (d+i-j)th symbol of the training sequence. a_i denotes the ith symbol of a sequence TS of 26 symbols:

$$A = \begin{array}{ccccc} a_4 & a_3 & a_2 & a_1 & a_0 \\ a_5 & a_4 & a_3 & a_2 & a_1 \\ a_6 & a_5 & a_4 & a_3 & a_2 \\ a_7 & \ldots & \ldots & \ldots & \\ \ldots & \ldots & \ldots & \ldots & \\ \ldots & \ldots & \ldots & \ldots & \\ a_{25} & \ldots & \ldots & \ldots & a_{21} \end{array}$$

The training sequence is chosen so that the matrix A^tA can be inverted, where the operator $.^t$ represents transposition.

In the receive signal S, the first four symbols s_0 to s_3 are ignored because, given that the dispersion of the channel is 4, they also depend on the unknown symbol transmitted before the training sequence. The receive signal is defined hereinafter as a vector S having for its components the received symbols s_4, s_5, s_6, ..., s_{25}.

The estimated impulse response X therefore takes the form:

$$X = (A^tA)^{-1}A^t.S$$

The next step of the method of the invention is to establish a statistic of this impulse response, where "statistic" means a data set reflecting the average value of the impulse response over an analysis period.

A matrix is therefore constructed for smoothing the various estimates X obtained during the analysis period to obtain an estimate of the covariance associated with

the impulse response. In this context the term "smoothing" is to be understood very broadly, i.e. as referring to any operation for smoothing or averaging the impulse response over the analysis period.

A first example of smoothing entails calculating the average of the matrix XX^h over the analysis period, which is assumed to include m training sequences:

$$L(XX^h) = \frac{1}{m}\sum_{1}^{m} XX^h$$

The operator $.^h$ represents the Hermitian transformation or complex conjugate transpose.

A second example of smoothing entails, after the ith training sequence is received, updating the smoothing matrix $L_{i-1}(XX^h)$ obtained on the (i-1)th training sequence by means of multiplier coefficient α, this factor generally being known as the smoothing forget factor and ranging from 0 to 1:

$$L_i(XX^h) = \alpha X_i X_i^h + (1-\alpha) L_{i-1}(XX^h)$$

Any means can be used for this initialization, including the first estimate X obtained, or an average obtained as described for a small number of training sequences.

For simplicity, the smoothing matrix $L(XX^h)$, which is in fact a statistical characterization matrix, is denoted L below.

The method then includes a step of looking for (eigenvalue, eigenvector) pairs of the characterization matrix.

This step is not described in more detail because it is well-known to the skilled person.

The eigenvalues λ_i are then classified in decreasing order. Their sum corresponds to the energy of the characterization signal X made up in part of a wanted signal which images the source signal and in part of the additive noise N of the transmission channel.

The dominant eigenvalues, the ones which are the

highest, represent the wanted signal, whereas the lowest eigenvalues represent noise.

A first option is to retain a predetermined number of dominant eigenvalues. For example, the first two eigenvalues λ_1 and λ_2 are retained for an impulse response with five coefficients.

A second option is to consider the wanted signal to have an energy which is a predetermined fraction f of the energy of the characterization signal. Thus using the notation λ_i for the eigenvalues with i varying from 1 to p, there will be d dominant eigenvalues, where d is obtained as follows:

$$\frac{\sum_{i=1}^{d} \lambda_i}{\sum_{i=1}^{p} \lambda_i} \le f \quad \text{and} \quad \frac{\sum_{i=1}^{d+1} \lambda_i}{\sum_{i=1}^{p} \lambda_i} > f$$

The fraction f can be fixed *a priori*, for example at 95%. This fraction can also be derived from the signal-to-noise ratio of the receive signal obtained elsewhere.

A third option, and doubtless that offering the best performance, is for the additive noise N to be estimated directly from the receive signal and from the measurement matrix A. If N_0 denotes the noise vector affecting the receive signal:

$$S = AX + N_0$$

Given that the vectors S and N_0 have 22 components, the additive noise N can be expressed as follows:

$$N = \left(\frac{1}{22}\right)(S - AX)^h\,(S - AX)$$

This estimate of the additive noise can naturally be averaged and smoothed.

Using the previous notation, and standardizing the energies:

$$\sum_{i=d}^{p} \lambda_i \geq N \quad \text{and} \quad \sum_{i=d+1}^{p} \lambda_i < N$$

Thus the dominant eigenvalues are obtained from a direct estimate of the noise.

Whatever option is previously adopted, the next step of the method consists of calculating the correlation function of the source signal with the sum of the eigenvectors v_i associated with the dominant eigenvalues λ_i.

The source signal is oversampled relative to the bit period and is therefore denoted g(t) where t represents time and is a discrete variable whose quantization increment is 1/32 bit period, for example. It is represented by a vector with the same number of dimensions as the characterization signal, i.e. five dimensions in this example. The correction function c(t) is calculated for t varying from -1 to +1 bit period, for example, using the following expression:

$$c(t) = \sum_{i=1}^{d} g(t).\, v_i$$

The period between the source signal g(t) and the eigenvector v_i represents a scalar product, in the conventional way.

The final step of the method looks for the value t_0 of t that is closest to zero, which corresponds to the first relative maximum of the correlation c(t). It is this particular value t_0 which gives the required synchronization error relative to the reception signal.

The following complementary function c'(t) can also be considered:

$$c'(t) = \sum_{i=d+1}^{p} g(t).\, v_i$$

Note that the particular value t_0 mentioned above can also be obtained by seeking the value of t closest to

zero, which corresponds to the first relative minimum of the complementary formation c'(t).

These two methods of obtaining the synchronization t_0 are therefore equivalent.

Turning to Figure 2, a second embodiment of the invention is considered in which the characterization signal is the receive signal S, so that the source signal is now the reference signal, or, in the case of the GSM, the training sequence TS when it is GMSK (Gaussian minimum shift keying) modulated.

The statistic of the characterization signal is therefore estimated by means of a characterization matrix which is now obtained by smoothing the various occurrences of the receive signal S. Once again, the term "smoothing" is to be understood in a very broad sense in this context.

The characterization matrix L therefore takes the following form:

$$L(SS^h) = \frac{1}{m}\sum_{1}^{m} SS^h$$

or

$$L_i(SS^h) = \alpha S_i S_i^h + (1-\alpha) L_{i-1}(SS^h)$$

The p' eigenvalues λ'_i of the matrix are then looked for, and as in the first embodiment the d' dominant eigenvalues are identified.

The correlation function f(t) of the modulated training sequence and the sum of the eigenvectors v'_i associated with the dominant eigenvalues λ'_i are now calculated.

Once again, the training sequence g'(t) is oversampled and is represented by a vector with 22 components. The correlation function therefore becomes:

$$f(t) = \sum_{i=1}^{d'} g'(t).v'_i$$

As before, a new complementary function f'(t) can be

defined:

$$f'(t) = \sum_{i=d'+1}^{p'} g'(t).v'_i$$

The method terminates in the same manner by seeking the first maximum of the correlation function f(t) or the first minimum of the complementary function f'(t).

The invention can therefore be implemented in various ways, the essential point being to have access to a source signal and of the result of transmitting it, i.e. the characterization signal.